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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	**Application No.** 10/790,070	**Applicant(s)** RHODES ET AL.	
	Examiner CHRISTINE DUONG	**Art Unit** 2416	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 04 November 2008.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1,3-6,46-58 and 60-65 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1,3-6,46-58 and 60-65 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Response to Amendment

This is in response to the Applicant's arguments and amendments filed on 04 November 2008 in which claim 59 is cancelled and claims 1, 3-6, 46-58, 60-65 are currently pending.

Claim Rejections - 35 USC § 102

1. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

2. Claims 49-53, 55 are rejected under 35 U.S.C. 102(e) as being anticipated by Shankar et al. (US Patent No. 6,570,869 B1 hereafter Shankar).

Regarding claim 49, Shankar discloses a method of choosing a call setup in a communication network (fig. 4).

The limitation, receiving a request from a calling telephone for call setup to a called telephone (“When a user initiates a voice call, a "setup" connection request message 402 is generated and ultimately transmitted by originating node 100 to originating coding unit 110” column 13 lines 12-17).

The limitation, determining whether the calling telephone and the called telephone support compatible voice compression algorithms (“The create connection

message 408 can also contain parameters that indicate the capabilities of the originating coding unit 110, for example, the encoding and compression types the originating coding unit 110 supports" column 14 lines 27-32 and "During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63).

The limitation, diverting the call to a data network if the called telephone supports a voice compression algorithm that is compatible with a voice compression algorithm of the calling telephone ("The connection message 418 contains the network address of the terminating coding unit 150 and the negotiated parameters or the common capabilities" column 15 lines 7-10 and "at this point, the voice call is active" column 15 lines 49).

Regarding claim 50, Shankar discloses everything claimed as applied above (see claim 49). In addition, Shankar discloses the limitation, determining whether the calling telephone and the called telephone have access to the same data network ("the terminating coding unit 150 establishes a bearer channel circuit 416 on the bearer packet-switching network 130. The bearer channel circuit 416 may be one-way (terminating coding unit 150 to originating coding unit 110) or two-way. If successful, the terminating coding unit 150 responds back with a connection message 418 to terminating signaling unit 140 over control link 154" (column 15 lines 1-7).

Regarding claim 51, Shankar discloses everything claimed as applied above (see claim 50). In addition, Shankar discloses the limitation, the data network is the Internet ("Packet-switching network 130 can be implemented as an IP network ... One example

of packet-switching network 130 is the global packet-switching network known as the Internet” column 4 lines 15-20).

Regarding claim 52, Shankar discloses everything claimed as applied above (see claim 49). In addition, Shankar discloses the limitation, the data network is the Internet (“Packet-switching network 130 can be implemented as an IP network ... One example of packet-switching network 130 is the global packet-switching network known as the Internet” column 4 lines 15-20).

Regarding claim 53, Shankar discloses a method (fig. 4).

The limitation, initiating a call setup between a calling party's audio device and a called party's audio device using a first path (“When a user initiates a voice call, a "setup" connection request message 402 is generated and ultimately transmitted by originating node 100 to originating coding unit 110” column 13 lines 12-17).

The limitation, before the call setup is completed, determining whether the called party's audio device supports a voice compression algorithm compatible with a voice compression algorithm supported by the calling party's audio device (“The create connection message 408 can also contain parameters that indicate the capabilities of the originating coding unit 110, for example, the encoding and compression types the originating coding unit 110 supports” column 14 lines 27-32 and “During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels” column 14 lines 61-63).

The limitation, based on the determination, completing the call setup using a second path different from the first path (“The connection message 418 contains the

network address of the terminating coding unit 150 and the negotiated parameters or the common capabilities" column 15 lines 7-10 and "at this point, the voice call is active" column 15 lines 49).

Regarding claim 55, Shankar discloses everything claimed as applied above (see claim 53). In addition, Shankar discloses the limitation, the second path includes a data network ("Packet-switching network 130 can be implemented as an IP network ... One example of packet-switching network 130 is the global packet-switching network known as the Internet" column 4 lines 15-20).

Claim Rejections - 35 USC § 103

3. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. Claims 1, 3, 5-6, 46-48, 54, 56-64 rejected under 35 U.S.C. 103(a) as being unpatentable over Shankar in view of Berry et al. (US Patent NO. 5,758,256 hereafter Berry).

Regarding claim 1, Shankar discloses a method (fig. 4).

The limitation, determining whether a called party's audio device is able to support at least one voice compression algorithm supported by a calling party's audio device ("The create connection message 408 can also contain parameters that indicate the capabilities of the originating coding unit 110, for example, the encoding and compression types the originating coding unit 110 supports" column 14 lines 27-32 and

“During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels” column 14 lines 61-63).

The limitation, exchanging voice signals between said called party's audio device and said calling party's audio device via a data network, if said called party's audio device is able to support said at least one voice compression algorithm (“The connection message 418 contains the network address of the terminating coding unit 150 and the negotiated parameters or the common capabilities" column 15 lines 7-10 and “at this point, the voice call is active” column 15 lines 49).

However, Shankar fails to specifically disclose the limitation, said determining step is accomplished by exchanging messages between said called party's audio device and said calling party's audio device via a circuit switched network.

Nevertheless, Berry teaches a PSTN 26 in fig. 1.

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to exchange messages between said called party's audio device and said calling party's audio device via a circuit switched network because “the packet-switching network 130 may even be overlaid on to the PSTN” (Shankar column 4 lines 17-18).

Regarding claim 3, Shankar and Berry discloses everything claimed as applied above (see claim 2). In addition, Shankar discloses the limitation, said messages are modified circuit control signaling messages (“common signaling protocol XISUP is an extension of Integrated Services Digital Network User Part (ISUP)” column 12 lines 3-5).

Regarding claim 5, Shankar and Berry discloses everything claimed as applied above (see claim 1). In addition, Shankar discloses the limitation, each of said audio devices is one of a wired telephone, a wireless telephone, and an Internet protocol (IP) based computer telephone ("Originating node 100 can be implemented as a Private Branch eXchange (PBX), a telephone switch, a "smart phone" capable of generating voice calls, a wireless PBX, or a legacy telecommunications system. Similarly, terminating node 160 can also be a PBX, telephone switch, telephone, a wireless PBX, or legacy telecommunications system" column 3 line 65 to column 4 line 3).

Regarding claim 6, Shankar and Berry discloses everything claimed as applied above (see claim 1). In addition, Shankar discloses the limitation, said data network is an Internet protocol (IP) network ("Packet-switching network 130 can be implemented as an IP network ... One example of packet-switching network 130 is the global packet-switching network known as the Internet" column 4 lines 15-20).

Regarding claim 46, Shankar discloses a method of setting up a voice call between two digital wireless telephones (fig. 4).

The limitation, receiving, at a switch in a communication network, a call setup request from a calling digital wireless phone ("When a user initiates a voice call, a "setup" connection request message 402 is generated and ultimately transmitted by originating node 100 to originating coding unit 110" column 13 lines 12-17).

The limitation, determining, prior to completing the call setup between the calling digital wireless telephone and a called digital wireless telephone, whether the called wireless digital telephone uses a voice compression algorithm that is compatible with a

voice compression algorithm used by the calling digital wireless telephone ("The create connection message 408 can also contain parameters that indicate the capabilities of the originating coding unit 110, for example, the encoding and compression types the originating coding unit 110 supports" column 14 lines 27-32 and "During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63).

The limitation, setting up the voice call over the Internet if the called digital wireless telephone and the calling digital wireless telephone use compatible voice compression algorithms ("The connection message 418 contains the network address of the terminating coding unit 150 and the negotiated parameters or the common capabilities" column 15 lines 7-10 and "at this point, the voice call is active" column 15 lines 49).

However, Shankar fails to specifically disclose digital wireless telephones and setting up the voice call over the communication network if the called digital wireless telephone and the calling digital wireless telephone use incompatible voice compression algorithms.

Nevertheless, Berry teaches "the origination or destination port may be a portable subscriber unit and usable in mobile telephone environments" (column 6 lines 30-32) and "if the user-user information indicates that a compatible FSU does not exist at the destination port 40, the source BSC 20 does not packetize the compressed digital signal. Instead, the source BSC 20 uses the TRF STM to decompress the voice information for transmission over PSTN lines" (column 6 lines 10-14).

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have digital wireless telephones and set up the voice call over the communication network if the called digital wireless telephone and the calling digital wireless telephone use incompatible voice compression algorithms because it will "minimize voice signal degradation and delay, as well as recognize the type of destination to which a voice signal is sent" (column 1 lines 41-43).

Regarding claim 47, Shankar and Berry discloses everything claimed as applied above (see claim 46). However, Shankar fails to specifically disclose the limitation, the communication network is a circuit switched network.

Nevertheless, Berry teaches a PSTN 26 in fig. 1.

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have the communication network be a circuit switched network because "the packet-switching network 130 may even be overlaid on to the PSTN" (Shankar column 4 lines 17-18).

Regarding claim 48, Shankar and Berry discloses everything claimed as applied above (see claim 47). However, Shankar fails to specifically disclose the limitation, the circuit switched network is a public switched telephone network.

Nevertheless, Berry teaches a PSTN 26 in fig. 1.

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have the circuit switched network be a public switched telephone network because "the packet-switching network 130 may even be overlaid on to the PSTN" (Shankar column 4 lines 17-18).

Regarding claim 54, Shankar discloses everything claimed as applied above (see claim 53). However, Shankar fails to specifically disclose the limitation, the first path includes a public switched telephone network.

Nevertheless, Berry teaches a PSTN 26 in fig. 1.

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have the communication network be a circuit switched network because “the packet-switching network 130 may even be overlaid on to the PSTN” (Shankar column 4 lines 17-18).

Regarding claim 56, Shankar discloses everything claimed as applied above (see claim 53). However, Shankar fails to specifically disclose the limitation, based on the determination, completing the call setup on the first path.

Nevertheless, Berry teaches “if the user-user information indicates that a compatible FSU does not exist at the destination port 40, the source BSC 20 does not packetize the compressed digital signal. Instead, the source BSC 20 uses the TRF STM to decompress the voice information for transmission over PSTN lines” (column 6 lines 10-14).

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to based on the determination, complete the call setup on the first path because it will "minimize voice signal degradation and delay, as well as recognize the type of destination to which a voice signal is sent” (column 1 lines 41-43).

Regarding claim 57, Shankar discloses a method (fig. 4).

The limitation, sending call setup signals via a circuit-switched network to set up a call between a calling party's telephone and a called party's telephone (“When a user initiates a voice call, a "setup" connection request message 402 is generated and ultimately transmitted by originating node 100 to originating coding unit 110” column 13 lines 12-17).

The limitation, consulting information relating to compatibility of respective voice compression algorithms supported by the calling party's telephone and the called party's telephone (“The create connection message 408 can also contain parameters that indicate the capabilities of the originating coding unit 110, for example, the encoding and compression types the originating coding unit 110 supports” column 14 lines 27-32 and “During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels” column 14 lines 61-63).

The limitation, based on the information, sending call setup signals via a data network to complete the call setup (“The connection message 418 contains the network address of the terminating coding unit 150 and the negotiated parameters or the common capabilities" column 15 lines 7-10 and “at this point, the voice call is active” column 15 lines 49).

The limitation, the information is included in a request issued by the calling party's telephone and in a response to the request from the called party's telephone (fig. 4 shows a call flow diagram between the originating node and terminating node).

However, Shankar fails to specifically disclose a circuit-switched network.

Nevertheless, Berry teaches a PSTN 26 in fig. 1.

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have a circuit-switched network because “the packet-switching network 130 may even be overlaid on to the PSTN” (Shankar column 4 lines 17-18).

Regarding claim 58, Shankar and Berry discloses everything claimed as applied above (see claim 57). In addition, Shankar discloses the limitation, the information further relates to whether both the calling party's telephone and the called party's telephone have access to the data network (“the terminating coding unit 150 establishes a bearer channel circuit 416 on the bearer packet-switching network 130. The bearer channel circuit 416 may be one-way (terminating coding unit 150 to originating coding unit 110) or two-way. If successful, the terminating coding unit 150 responds back with a connection message 418 to terminating signaling unit 140 over control link 154” (column 15 lines 1-7).

Regarding claim 60, Shankar and Berry discloses everything claimed as applied above (see claim 57). In addition, Shankar discloses the limitation, the information is included in a network node (fig. 4 shows information originating and terminating from respective node).

Regarding claim 61, Shankar discloses a method (fig. 4).

The limitation, sending a call request message to a called digital wireless telephone via a public switched telephone network as a portion of a call setup procedure, the call request message including a list of voice compression algorithms

supported by a calling digital wireless telephone ("The create connection message 408 can also contain parameters that indicate the capabilities of the originating coding unit 110, for example, the encoding and compression types the originating coding unit 110 supports" column 14 lines 27-32 and "During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63).

The limitation, receiving a response message indicating whether the called digital wireless telephone can support one of the voice compression algorithms on the list, and whether the called digital wireless telephone can access a data network also accessible to the calling digital wireless telephone ("During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63 and "the terminating coding unit 150 establishes a bearer channel circuit 416 on the bearer packet-switching network 130. The bearer channel circuit 416 may be one-way (terminating coding unit 150 to originating coding unit 110) or two-way. If successful, the terminating coding unit 150 responds back with a connection message 418 to terminating signaling unit 140 over control link 154" (column 15 lines 1-7).

The limitation, if the response message indicates that the called digital wireless telephone can support one of the voice compression algorithms on the list and that the called digital wireless telephone can access the data network, completing the call setup procedure via the data network ("The connection message 418 contains the network address of the terminating coding unit 150 and the negotiated parameters or the

common capabilities" column 15 lines 7-10 and “at this point, the voice call is active” column 15 lines 49).

However, Shankar fails to specifically disclose digital wireless telephones, a public switched telephone network and if the response message indicates that the called digital wireless telephone cannot support one of the voice compression algorithms on the list or that the called digital wireless telephone cannot access the data network, completing the call setup procedure via the public switched telephone network.

Nevertheless, Berry teaches “the origination or destination port may be a portable subscriber unit and usable in mobile telephone environments” (column 6 lines 30-32), a PSTN 26 in fig. 1 and “if the user-user information indicates that a compatible FSU does not exist at the destination port 40, the source BSC 20 does not packetize the compressed digital signal. Instead, the source BSC 20 uses the TRF STM to decompress the voice information for transmission over PSTN lines” (column 6 lines 10-14).

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have digital wireless telephones, a public switched telephone network and if the response message indicates that the called digital wireless telephone cannot support one of the voice compression algorithms on the list or that the called digital wireless telephone cannot access the data network, completing the call setup procedure via the public switched telephone network because it will "minimize voice signal degradation and delay, as well as recognize the type of destination to which a voice signal is sent” (column 1 lines 41-43).

Regarding claim 62, Shankar and Berry discloses everything claimed as applied above (see claim 61). In addition, Shankar discloses the limitation, the data network is the Internet ("Packet-switching network 130 can be implemented as an IP network ... One example of packet-switching network 130 is the global packet-switching network known as the Internet" column 4 lines 15-20).

Regarding claim 63, Shankar discloses a method (fig. 4).

The limitation, performing a phase of a call setup procedure using a communication path that includes a mobile switching center (MSC) coupled to a data network (packet-switching network 130) and a public switched telephone network (PSTN), the phase including sending a call request message received from a calling digital wireless telephone at the MSC to a called digital wireless telephone via the PSTN, the call request message including a list of voice compression algorithms supported by the calling digital wireless telephone ("The create connection message 408 can also contain parameters that indicate the capabilities of the originating coding unit 110, for example, the encoding and compression types the originating coding unit 110 supports" column 14 lines 27-32 and "During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63).

The limitation, determining, based on a response message from the called digital wireless telephone, whether the called digital wireless telephone supports one of the voice compression algorithms on the list and whether the called digital wireless telephone has access to the data network ("During the negotiation session 414, the

originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63 and "the terminating coding unit 150 establishes a bearer channel circuit 416 on the bearer packet-switching network 130. The bearer channel circuit 416 may be one-way (terminating coding unit 150 to originating coding unit 110) or two-way. If successful, the terminating coding unit 150 responds back with a connection message 418 to terminating signaling unit 140 over control link 154" (column 15 lines 1-7).

The limitation, if the called digital wireless telephone supports one of the voice compression algorithms on the list and has access to the data network, completing the call setup procedure via the data network ("The connection message 418 contains the network address of the terminating coding unit 150 and the negotiated parameters or the common capabilities" column 15 lines 7-10 and "at this point, the voice call is active" column 15 lines 49).

However, Shankar fails to specifically disclose a MSC coupled to a PSTN, digital wireless telephone, sending a call request message received from a calling digital wireless telephone at the MSC to a called digital wireless telephone via the PSTN and if the called digital wireless telephone does not support one of the voice compression algorithms on the list or does not have access to the data network, completing the call setup procedure via the PSTN.

Nevertheless, Berry teaches a switch 25 and PSTN 26 in fig. 1 where "the switch is preferably a mobile switching center (MSC) with user-user signaling capability" (column 2 lines 65-67), "the origination or destination port may be a portable subscriber

unit and usable in mobile telephone environments" (column 6 lines 30-32), "when a telephone call on the fixed wireless cellular system is initiated, the source BSC 20 exchanges call setup information with the destination BSC 30" (column 4 lines 24-27), and "if the user-user information indicates that a compatible FSU does not exist at the destination port 40, the source BSC 20 does not packetize the compressed digital signal. Instead, the source BSC 20 uses the TRF STM to decompress the voice information for transmission over PSTN lines" (column 6 lines 10-14).

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to have a MSC coupled to a PSTN, digital wireless telephone, send a call request message received from a calling digital wireless telephone at the MSC to a called digital wireless telephone via the PSTN and if the called digital wireless telephone does not support one of the voice compression algorithms on the list or does not have access to the data network, complete the call setup procedure via the PSTN because it will "minimize voice signal degradation and delay, as well as recognize the type of destination to which a voice signal is sent" (column 1 lines 41-43).

Regarding claim 64, Shankar and Berry discloses everything claimed as applied above (see claim 63). In addition, Shankar discloses the limitation, the data network is the Internet ("Packet-switching network 130 can be implemented as an IP network ... One example of packet-switching network 130 is the global packet-switching network known as the Internet" column 4 lines 15-20).

5. Claim 65 is rejected under 35 U.S.C. 103(a) as being unpatentable over Shankar in view of Farris (US Patent No. 6,064,653).

It is noted, with respect to claim 65, that the language used by Applicant merely suggests or makes optional those features described as “adapted to”; such language does not require steps to be performed nor limits the claim to a particular structure. It has been held that the recitation that an element is “adapted to” perform a function is not a positive limitation but only requires the ability to so perform. It does not constitute a limitation in any patentable sense. In re Hutchinson, 69 USPQ 138.

Regarding claim 65, Shankar discloses a method for diverting an Integrated Services Digital Network User Part (ISUP) network talkpath to a data network talkpath (fig. 4).

The limitation, determining, whether a called party's telephone is adapted to exchange voice signals via a same data network to which a calling party's telephone is adapted to exchange voice signals (“The create connection message 408 can also contain parameters that indicate the capabilities of the originating coding unit 110, for example, the encoding and compression types the originating coding unit 110 supports” column 14 lines 27-32 and “the terminating coding unit 150 establishes a bearer channel circuit 416 on the bearer packet-switching network 130. The bearer channel circuit 416 may be one-way (terminating coding unit 150 to originating coding unit 110) or two-way. If successful, the terminating coding unit 150 responds back with a connection message 418 to terminating signaling unit 140 over control link 154” (column 15 lines 1-7).

The limitation, establishing the data network talkpath using resources associated with said same data network if said called party's telephone is adapted to exchange voice signals via said same data network ("During the negotiation session 414, the originating coding unit 110 and the terminating coding unit 150 negotiate appropriate compression and decoding levels" column 14 lines 61-63).

The limitation, exchanging voice signals between said called party's telephone and said calling party's telephone using the data network talkpath ("The connection message 418 contains the network address of the terminating coding unit 150 and the negotiated parameters or the common capabilities" column 15 lines 7-10 and "at this point, the voice call is active" column 15 lines 49).

However, Shankar fails to specifically disclose using an ISUP signaling path and ISUP signaling path being established during a process of establishing the ISUP network talkpath.

Nevertheless, Farris teaches "the control part of SS7 protocol is known as Integrated Services Digital Network User Part (ISUP). ISUP determines the procedures for setting up, coordinating, and taking down trunk calls on the SS7 network" (column 6 lines 55-59).

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to use an ISUP signaling path and ISUP signaling path being established during a process of establishing the ISUP network talkpath because "signaling between switching offices is required for transmitting routing and destination information, for transmitting alerting messages such as to indicate the arrival

of an incoming call, and for transmitting supervisor information, e.g. relating to line status" (column 6 lines 60-64).

6. Claim 4 is rejected under 35 U.S.C. 103(a) as being unpatentable over Shankar and Berry further in view of Wilson (US Patent No. 6,169,734 B1).

Regarding claim 4, Shankar and Berry discloses everything claimed as applied above (see claim 1). However, Shankar and Berry fails to specifically disclose compressing said voice signals using said at least one voice compression algorithm at one of said called party's audio device and said calling party's audio device, sending said compressed voice signals to one other of said called party's audio device and said calling party's audio device via said data network and decompressing said compressed voice signals using said at least one voice compression algorithm at said one other of said called party's audio device and said calling party's audio device.

Nevertheless, Wilson teaches "allow users to receive and transmit compressed Internet voice messages across the Internet. Typically, a user at one end of the connection speaks into a microphone attached to a Personal Computer ("PC"). The microphone carries the audio voice signal to a processor board in the PC which digitizes the signal and creates a digital voice file. The voice file is then typically compressed and transferred to a selected recipient at a distant point on the Internet. Once received, the voice file is decompressed and converted via digital signal processing to an audible signal intelligible to the human ear" (column 1 lines 32-42).

Therefore, it would have been obvious to a person having ordinary skill in the art at the time the invention was made to compress said voice signals using said at least

one voice compression algorithm at one of said called party's audio device and said calling party's audio device, send said compressed voice signals to one other of said called party's audio device and said calling party's audio device via said data network and decompress said compressed voice signals using said at least one voice compression algorithm at said one other of said called party's audio device and said calling party's audio device because “the device can combine the simplicity of operation of the POTS with low cost audio access to the Internet” (column 1 lines 61-62).

Response to Arguments

7. Applicant's arguments have been fully considered but they are not persuasive.

Applicants have argued regarding claim 1 that “the cited portion of Shankar refers to terminating coding unit (element 150) and originating coding unit (element 110). These are not “said called party’s audio device” and “calling party’s audio device” (page 8).

In response to Applicants’ argument, the examiner respectfully disagrees. The calling party's audio has been interpreted with the following elements either alone or in combination of originating node 100, originating coding unit 110 and originating signaling unit 120 and the called party's audio has been interpreted with the following elements either alone or in combination of terminating node 160, terminating coding unit 150 and terminating signaling unit 140. Therefore, Shankar discloses said called party’s audio device and calling party’s audio device.

Applicants have argued regarding claim 1 that "the originating node 100 and terminating node 160 of Shankar are not exchanging messages to determine a network to use" (page 8).

In response to Applicants' argument, the examiner respectfully disagrees. Shankar discloses in fig. 4 that an originating node 100 (along with its coding unit 110 and signaling unit 120) and terminating node 160 (along with its coding unit 150 and signaling unit 140) exchange information to negotiate a connection and compression capabilities. Therefore, Shankar discloses the originating node 100 and terminating node 160 are exchanging messages to determine a network to use.

Applicants have argued regarding claim 1 that "Shankar states that the coding unit and the terminating unit negotiate appropriate compression and decoding levels – not the originating node 100 and the terminating node 160" (page 8).

In response to Applicants' argument, the examiner respectfully disagrees. Shankar discloses in fig. 4 that an originating node 100 (along with its coding unit 110 and signaling unit 120) and terminating node 160 (along with its coding unit 150 and signaling unit 140) exchange information to negotiate a connection and compression capabilities. Therefore, Shankar discloses the originating node 100 and the terminating node 160 negotiate compression.

Applicants have argued regarding claims 1, 46, 57, 61, 63, 65, 49, 53 that "nowhere in Shankar is there a teaching or suggestion of using a first network and then

switching to a second network based on support of a compression algorithm" (pages 8, 9, 10, 11, 12, 13).

In response to Applicants' argument, the examiner respectfully disagrees. Shankar discloses "After the originating signaling unit 120 has performed this call routing capability, the originating signaling unit 120 transmits a signaling message, including information for establishing the voice and the network address of the originating coding unit 110, through network 132 to terminating signaling unit 140" (Shankar column 5 lines 47-52) and "After performing the bearer selection, the terminating signaling unit 140 controls the terminating coding unit 150 to establish a bearer channel for the voice through packet-switching network 130" (Shankar column 5 lines 55-59). This shows that the network 132 is used to setup a connection and then the packet switched network 130 is used to connect the call. Therefore, Shankar discloses using a first network and then switching to a second network based on support of a compression algorithm.

In response to applicant's arguments regarding claims 1, 57 (page 9, 10) against the references individually ("in Berry the PSTN is not being used by two audio devices to determine whether the audio devices can switch to a data network, if said called party's audio device is able to support said at least one voice compression algorithm"), one cannot show nonobviousness by attacking references individually where the rejections are based on combinations of references. See *In re Keller*, 642 F.2d 413,

208 USPQ 871 (CCPA 1981); *In re Merck & Co.,* 800 F.2d 1091, 231 USPQ 375 (Fed. Cir. 1986).

Applicants have argued regarding claim 57 that “nor does Shankar teach or suggest switching to a second network (data network) based on information received from the called party’s telephone” (page 10).

In response to Applicants’ argument, the examiner respectfully disagrees. Shankar discloses “a coding unit is controlled by a signaling unit, for example, to establish a bearer channel for the voice data over the packet-switching network 130” (Shankar column 5 lines 28-31) and “After the originating signaling unit 120 has performed this call routing capability, the originating signaling unit 120 transmits a signaling message, including information for establishing the voice and the network address of the originating coding unit 110, through network 132 to terminating signaling unit 140” (Shankar column 5 lines 47-52). This shows that the voice is transmitted through packet switching network 130 after negotiation information is transmitted from network 132. Therefore, Shankar discloses switching to a second network (data network) based on information received from the called party’s telephone.

In response to applicant's argument regarding claim 65 that the references fail to show certain features of applicant’s invention, it is noted that the features upon which applicant relies (i.e., “exchanging voice signals using a data network if the called party’s telephone is adapted to exchange voice signals via said same data network otherwise using an ISUP signaling path”) are not recited in the rejected claim(s). Although the

claims are interpreted in light of the specification, limitations from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993).

Applicants have argued regarding claim 53 that “nor does Shankar teach or suggest completing a call setup using a second path different from a first path based on a determination of support of a compression algorithm of a called party's audio device” (page 13).

In response to Applicants’ argument, the examiner respectfully disagrees. Shankar discloses “a coding unit is controlled by a signaling unit, for example, to establish a bearer channel for the voice data over the packet-switching network 130” (Shankar column 5 lines 28-31) and “After the originating signaling unit 120 has performed this call routing capability, the originating signaling unit 120 transmits a signaling message, including information for establishing the voice and the network address of the originating coding unit 110, through network 132 to terminating signaling unit 140” (Shankar column 5 lines 47-52) and “the MDCX message 426 also contains the parameters negotiated between the originating coding unit 110 and the terminating coding unit 150” (Shankar column 15 lines 20-24). This shows that the voice is transmitted through packet switching network 130 after negotiation information regarding compression is transmitted from network 132. Therefore, Shankar discloses completing a call setup using a second path different from a first path based on a determination of support of a compression algorithm of a called party's audio device.

Conclusion

8. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to CHRISTINE DUONG whose telephone number is (571)270-1664. The examiner can normally be reached on Monday - Friday: 830 AM-6 PM EST with first Friday off.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Seema Rao can be reached on (571) 272-3174. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/Kevin C. Harper/
Primary Examiner, Art Unit 2416

/Christine Duong/
Examiner, Art Unit 2416
01/12/2009